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13. ABSTRACT (Maximum 200 words)

We addressed the problem of resolving signals that are closely spaced in frequency using short data segments. Such problems are of interest in electronic warfare and for locating the sources of electromagnetic radiation such as radar and communication devices. Effective, computationally efficient methods of high resolution parameter estimation were developed. The performance of the algorithms were predicted by theoretical results and verified by computer simulations. We also addressed the problem of tracking signals whose spectrum was varying with time. Signals were assumed to consist of multiple time-varying signal components and improved methods of residual signal analysis were developed to isolate and track the individual components. The methods were applied to real world signals such as voiced speech to decompose them into their harmonic components. During this period two PhD theses were completed. The research results were published in the form of ten conference publications and six journal publications.

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Abstract

We addressed the problem of resolving signals that are closely spaced in frequency using short data segments. Such problems are of interest in electronic warfare and for locating the sources of electromagnetic radiation such as radar and communication devices. Effective, computationally efficient methods of high resolution parameter estimation were developed. The performance of the algorithms were predicted by theoretical results and verified by computer simulations. We also addressed the problem of tracking signals whose spectrum was varying with time. Signals were assumed to consist of multiple time-varying signal components and improved methods of residual signal analysis were developed to isolate and track the individual components. The methods were applied to real world signals such as voiced speech to decompose them into their harmonic components. During this period two PhD theses were completed. The research results were published in the form of ten conference publications and six journal publications.

Keywords: Signal Analysis, Non-Stationary Signal Decomposition,
Speech Analysis, Co-Channel Interference Mitigation

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1 Executive Summary

During the period of this contract we have addressed the problem of resolving signals that are closely spaced in frequency using short data segments. Such problems are of interest in electronic warfare and for locating the sources of electromagnetic radiation such as radar and communication devices. Effective computationally efficient methods of high resolution parameter estimation were developed. The performance of the algorithms were predicted by theoretical results and verified by computer simulations. We also addressed the problem of tracking signals whose spectrum was varying with time. Signals were assumed to consist of multiple components and improved methods of residual signal analysis were developed. The methods were applied to real world signals such as voiced speech signals to decompose them into their harmonic components. During this period two PhD theses were completed. The research results were published in the form of ten conference publications and six journal publications.

2 Accomplishments

2.1 Completed PhD Dissertations;

The following two students have completed their PhD dissertations and were partly or fully supported by this AFOSR Contract. The Dissertation Abstracts are given below, A copy of the entire dissertation can be obtained from the Principal Investigator if required.

Dissertation Title: Analysis of Non-Stationary, Multi-Component Signals with Applications to Speech
Student's Name: C.S.Ramalingam
Adviser: R.Kumaresan

ABSTRACT

In this thesis we address the problem of decomposing a signal containing multiple non-stationary sinusoids into its components. A non-stationary sinusoid is one whose envelope and frequency are functions of time; we assume that these variations are slow enough so that the components occupy distinct regions in the spectrum at each instant of time. We propose to separate the components by using moving bandpass filters. Each moving bandpass filter is implemented as a cascade of demodulator, low-pass filter, and remodulator. We use the instantaneous frequency of the assigned component for tracking. We point out that while the raw instantaneous frequency may be potentially misleading, its smoothed counterpart is physically very meaningful and can be reliably used for moving the filters. The burden on a filter to isolate its component is eased by subtracting from its input an estimate of other signal components. This is based on the principle of Residual Signal Analysis, originally proposed by Costas. Our procedure results in a cleaner separation of the components when compared with his implementation. We have also proposed a feedforward structure to carry out Residual Signal Analysis. Using these methods for voiced speech analysis we have found that the components need not necessarily be precisely harmonic, but only nominally so, and that there can be brief periods of time when some of them can become significantly mistuned; the envelopes of some of the components displayed significant amplitue modulation. Teager has raised a number of objections against the popularly used source-filter model and has claimed that it is completely wrong. Our simulation results indicate that this model reproduced quite acurately the frequency mistuning seen in the examples of natural speech we have looked at. We also demonstrate that the polynomial envelope and phase model for non-stationary signal analysis is sensitive to model mismatch and does not automatically yield the increased accuracy expected of it. Further, the recently proposed DESA-la algorithm of Maragos, et al. is shown to be equivalent to Prony's method for the noiseless sinusoid case and an ad hoc variant of it in the presence of noise.

Dissertation Title: Efficient Methods for High-Resolution Parameter Estimation and Ambiguity Resolution
Student's Name: H.Ge
Adviser: D.W.Tufts

ABSTRACT

This work presents some efficient, effective, new methods for high resolution parameter estimation. Relatively small amounts of simple computations are needed, yet these methods still give quite satisfactory results compared to traditional methods. The computational and hardware simplicity of these high resolution approaches is achieved at the cost of possible inherent ambiguities. Therefore, another aspect of this work is to include methods to resolve ambiguities. Certain probabilities are defined in order to evaluate the ability of the methods to resolve ambiguities. The theoretical results are verified thourgh computer simulations.

Manuscript 1 presents some advances in phase-only signal processing in estimating frequencies of sinusoids. Possible ambiguity is resolved by the use of envelope-data. Manuscript 2 also deals with the problem of estimating frequencies of sinusoids. A computationally simple Sparse Linear Prediction (SLP) method is proposed to estimate the frequencies of sinusoids. The possible ambiguities are resolved by properly choosing values of delays and by introducing an additional parallel computation. Extension of the proposed SLP method to the case of under-sampled time series data is also provided. Manuscript 3 applies the idea in Manuscript 2 to estimate the spatial frequencies or directions of arrival of source signals from the outputs of a sparse linear array. The final part of this work provides the justifications and mathematical proofs of the arguments use throughout this work.

2.2 Research Completed and Published

We addressed the problem of decomposing a given composite signal into its multiple non-stationary components is being studied. Its application lies in areas such as Speech Recognition and Communications. The research conducted till date are briefly itemized below.

- RISC Algorithm for tracking multiple non-stationary signals
  The Residual Interfering Signal Canceler (RISC) procedure is based on the principles of Residual-Signal-Analysis (RSA). The algorithm developed was an improvement over Costas's estimator-predictor filter-bank structure. It was used to successfully track multiple, synthetic amplitude and frequency modulated (referred to as AM-FM) signals. The procedure was also applied to retrieve the nominally harmonic partials present in real voiced speech signals. This research work was published in the Proceedings of the *Seventh SSAP Workshop*, Quèbec City, Canada, Jun. 1994, pp. 207–210.

- A New version of RISC applicable to real-valued signals
  This new version was an extension of the RISC procedure. It could be applied directly to track the non-stationary components present in a real-valued multi-component signal; the previous RISC method was applicable only to complex data. This reaesrch was published in the Proceedings of the *IEEE-SP International Symposium on Time-Frequency and Time-Scale Analysis*, Oct. 25-28, 1994, PA, pp 500–503.

- A Dynamic Tracking Filter Bank Procedure For Multicomponent Signal Decomposition
  An algorithm that employed multiple Dynamic Tracking Filters (DTFs) in a filter-bank structure was developed. Although it was based on the same RSA principles, the resulting procedure could achieve clean separation of non-stationary components without resorting to the prediction/subtraction type techniques; the later methods have problems when the number of components increase. The ability of the algorithm to simultaneously track the harmonics as well as the formants present in real voiced speech signals (taken from the TIMIT database) was demonstrated. The work was published in the Proceedings of the *IEEE International Conference on Acoustics, Speech and Signal Processing*, May. 8–12, 1995, Detroit, Michigan.

- Modeling of Cochlear Functions in the Human Auditory System
  Although this on-going research is still in a very rudimentary stage, an attempt was made to model the signal processing performed by the Cochlea or the auditory periphery. A theory was proposed that an acoustic waveform, impinging on the human ear, is transformed to a Minimum/Maximum Phase signal. Since the envelope and the phase of such a signal carries redundant information, the hypothesis was that it is further represented by zero-crossings alone. This research was published in the *Twenty-Eighth Annual Asilomar Conference on Signals, Systems, and Computers*, Oct. 31–Nov. 2 1994, Pacific Grove, California, pp 404–408.

- Application of RISC to Co-Channel Interference Mitigation
  The RSA based algorithms were applied to the problem of co-channel interference mitigation. This problem is typically encountered in the following areas: (1) In military communications there is a need to communicate in the presence of one or more intentional jammers, (2) In Civilian Communications, it is desired to demodulate an FM signal which is present in a crowded spectral environment (3) In Cellular Communications, there are one

or more interfering FM signals in addition to the signal-of-interest. In the process of applying our algorithms to tackle this problem, a detailed performance analysis of our procedures was carried out. These results will be submitted to the *IEE transactions on Communications*.

- Polynomial-Phase-Modeling (PPM) for Co-Channel Signal Separation

  The parametric-model-fitting methods were studied. An algorithm to demodulate an FM signal based on the PPM approach was developed. Closed-form expressions of the Cramer-Rao Lower Bounds for the phase, the instantaneous frequency and the frequency rate of a second-degree chirp signal were derived. The algorithms performance was evaluated in comparison with the CR-bounds. The problem of demodulating co-channel FM signals was addressed next. Two simple models were used to test the procedure. In one, a short segment of a two-component signal was modeled as a sum of two sinewaves. The frequencies of the sinusoids were estimated by minimizing a squared model-fitting error. By repeating the procedure for consecutive overlapping segments, the modulating signals were re-synthesized. In the second model, a longer segment was fit with two sinewaves whose frequencies changed linearly with time. The same procedure of resynthesizing the modulating signals was used. The performances of both the procedures were evaluated by applying them to a real-world Ham-Radio recorded co-channel FM signal mixture. These results will also be submitted to the *IEE transactions on Communications*.

2.3   Research in Progress:

- PCMA: A Parametric Constant Modulus Algorithm For Co-Channel Interference Mitigation

  Recently, a new algorithm called the PCMA was developed. It incorporates *a priori* knowledge of the modulating signal and the modulated carrier in a sophisticated parametric-model. The model parameters are estimated by minimizing a constant-modulus-error (CME) criterion. A gradient-descent algorithm to minimize the CME was developed. The procedure is an extension of the well-known Constant-Modulus-Algorithm (The CMA). Clean separation of synthetic as well as voice-modulated FM signals was achieved by the PCMA approach. We are currently studying the PCMA's limitations. The details of the procedure appeared in the proceedings of the *Twenty-Eighth Annual Asilomar Conference on Signals, Systems, and Computers*, Oct. 29–Nov. 1 1995, Pacific Grove, California.

- New Signal Representations

  We are currently investigating new ways of representing signals. Specifically, we are interested in decomposing voiced speech signals represented as a linear combination of many Minimum/Maximum-phase components. Our ultimate goal is to extract the invariant modulating features, hopefully present in speech signals. Some preliminary research findings appeared in the proceedings of the *Twenty-Ninth Annual Asilomar Conference on Signals, Systems, and Computers*, Oct. 29–Nov. 1 1995, Pacific Grove, California.

- Modeling the Auditory Periphery of the American Bull-frog

  This reasearch is not directly connected with the AFOSR related research but has some tangential connections to it. In collaboration with the Neuroscience department of Brown University, a current investigation of the signal analysis performed by the front-end of the frog's auditory system is being conducted. Data corresponding to the neural spike train output (from the frog's eight nerve) is being collected by researchers at the Hunter laboratory of Brown University. Given the input stimulli and the nerve output, our primary intention is to model the functions of the frog's intermediate auditory pathway.

- **Speech source localization algorithm** A number of source localization or direction of arrival (DOA) finding algorithms are available for narrow band signals. However, speech signal can not be considered narrow banded. In this study, we propose an algorithm that will convert the voiced speech signal into a narrow band signal and then use MUSIC algorithm to find the DOA of the speech source. The proposed algorithm is based on the fact that voiced speech signal has an harmonic structure in its frequency components. Thus we can convert the frequency component at a higher harmonic component (nF_0) to that at the pitch (fundamental) frequency (F_0) once the pitch frequency is known. The advantage of using this algorithm over one that applies MUSIC to a single harmonic component is that it enhance the SNR in the resulting narrow band signal in two ways — frequency compression and signal averaging. The former refers to the procedure of adding (compressing) the energy at different harmonic components (nF_0, for $n = 1, 2, ...$) together; and the latter is the procedure of estimating the signal by averaging a few (say three) pitch periods of the original waveform prior to the frequency compression.

  The preliminary result of this study is accepted for presentation in the ASILOMAR-95 and for publication in the conference proceedings. The full report will be submitted to IEEE transaction for audio and speech processing.

- **A variable frame pitch determination algorithm** The above project requires a reliable pitch determination algorithm (PDA), preferably one that can take advantage of multiple channel inputs. The present aims to supply such a PDA. The proposed algorithm is based on the maximum likelihood (ML) PDA studied by a number of previous investigators (for example, Noll, 1969; Wise *et. a l*, 1974). However, the previous ML pitch estimators

are fixed frame PDAs. A speaker independent PDA must be able to estimate pitch periods ranging from about 2 msec to 20 msec. Accordingly, these PDAs must have a frame length of at least 40 msec (to be twice the longest potential period). Thus in the extreme case when pitch period is about 2msec, as many as 20 pitch periods are covered in the data frame. For this reason, the estimate accuracy and time resolution of the PDAs are severely compromized for high pitch frequency speech, especially during the begining and the end of a voiced speech segment. On the other hand, these PDAs suffer from inadequate frame length for low pitch frequency speech. In practice, the error rate seems lowest when frame length is about three times the true pitch period[?]. Hence, "A fixed frame PDA runs non-optimaly for most situations"[?].

The proposed variable frame PDA overcomes the above problem. The preliminary testing of this algorithm is very encouraging. This study was presented in the ICASSP-96 and is published in the conference proceedings. The full report will be submitted to IEEE transaction for audio and speech processing.

- **Performance comparison of some promising pitch determination algorithms** This study aims to select the best pitch determination algorithm for use in the first project. We have picked a few promising PDAs (Medan, *et. al*, 1991; Hermes, 1988; Wise *et. al*, 1974; Noll, 1969) and completed coding of these algorithms. These algorithms and the one proposed in the second project above will be compared and the result will be submitted to IEEE transaction for audio and speech processing.

3   Personnel Supported

- R.Kumaresan (Principal Investigator)
- D.W.Tufts (Faculty Associate)
- C.S.Ramalingam (Graduate Student, Completed his PhD Degree in Feb.1995)
- Hongya Ge (Graduate Student, Completed her PhD Degree in April 1995)
- Ashwin Rao (Graduate Student, Currently Enrolled in PhD Program)
- Xiaoshu Qian (Graduate Student, Currently Enrolled in PhD Program)

4   Publications

The research for the following publications were supported by the AFOSR Contract and this fact is mentioned in the acknowledgement section of these publicatons.

4.1   Book Chapters

R.Kumaresan, "Spectrum Analysis", Chapter 16, Handbook on Digital Signal Processing, John Wiley Inc., New York, NY, 1994.

4.2   Journal Publications

- "An Invariance Property of the Determinant of a matrix Whose elements are a sum of sinusoids and its application", R.Kumaresan and A.Sadasiv, Archive fur Elektronik und Uebertragungstechnik, vol.47, No.2, pp.119-122, 1993.
- "Multi-Dimensional System Identification from Impulse Response Data", Circuits, Systems and Signal Processing - Special Issue on Multivariable Systems, Vol.13, No.6, pp.759-782, Dec.1994.
- "Non-Stationary Signal Analysis With Applications to Voiced Speech", C.S.Ramalingam and R.Kumaresan submitted to *Journal of the Acoustical Society of America*, May 1995.
- "Effect of Model Mismatch on Polynomial Envelope and Phase Modeling of Non-stationary Sinusoids" C.S.Ramalingam and R.Kumaresan, Submitted to the IEE Transcation on Radar and Signal Processing, August, 1995.
- "Simple, Robust Digital Frequency Modulation", D.W.Tufts and H.Ge, Submitted to IEEE Transactions on Communications, 2/13/95.
- "On the Equivalence of DESA-1a and Prony's Method when the signal is a sinusoid", (C.S.Ramalingam), Submitted to the IEEE Signal Processing letters, Feb.1995.
- "Cramer-Rao Lower bounds on the Errors in Estimating Times of Arrival and Frequency of a Filtered Burst of Sinusoid", (H.Ge and D.W.Tufts), Accepted for Publication by the IEEE Trans. on Aerospace and Electronic Systems.

5 Interactions/Transitions

5.1 Participation in Conferences, Meetings etc.

The following papers were presented at various Signal processing Conferences and explicitly mention AFOSR support in the acknowledgements.

- "Effect of Model Mismatch on Polynomial Envelope and Phase Modeling of Non-stationary Sinusoids" C.S.Ramalingam and R.Kumaresan, in the Proceedings of the *IEEE International Conference on Acoustics, Speech and Signal Processing*, May. 9–12, 1995, Detroit, Michigan, USA, pp.1780-1783.

- " Dynamic Tracking Filters For Decomposing Nonstationary Sinusoidal Signals", A.Rao and R.Kumaresan, in the Proceedings of the *IEEE International Conference on Acoustics, Speech and Signal Processing*, May. 9–12, 1995, Detroit, Michigan, USA, pp.917-920.

- "Digital Estimation of Frequencies of Sinusoids from Wideband Undersampled Data" (H.Ge and D.W.Tufts) in the Proceedings of the *IEEE International Conference on Acoustics, Speech and Signal Processing*, May. 9–12, 1995, Detroit, Michigan, USA, pp.3110-3113.

- "A model for signal representation at the output of the cochlea based on minimum phase signals", A.Rao and R.Kumaresan, in the Proceedings of the *Twenty-Eighth Annual Asilomar Conference on Signals, Systems, and Computers*, Oct. 31–Nov. 2 1994, Pacific Grove, California, USA, pp 404–408

- "Estimation of the Spatial and Temporal Frequencies of Multiple Sinusoids using a Sparse Linear Array", H.Ge, D.W.Tufts and R.Kumaresan, in the Proceedings of the *Twenty-Eighth Annual Asilomar Conference on Signals, Systems, and Computers*, Oct. 31–Nov. 2 1994, Pacific Grove, California, USA, pp 821–824.

- "Time-Frequency Analysis Using The Residual Interference Signal Canceler Filter Bank", C.S.Ramalingam, A.Rao and R.Kumaresan, in the Proceedings of the *IEEE-SP International Symposium on Time- Frequency and Time-Scale Analysis*, Oct. 25-28, 1994, PA, USA, pp 500–503.

- "RISC: An improved Costas estimator-predictor filter-band for decomposing multi-component signals", R.Kumaresan, C.S.Ramalingam, and A.Rao in the Proceedings of the *Seventh SSAP Workshop*, Quèbec City, Canada, Jun. 1994, pp. 207–210.

- "PCMA: Parametric Constant Modulus Algorithm For Co-Channel Interference Mitigation", to appear in the Proceedings of the *Twenty-Ninth Annual Asilomar Conference on Signals, Systems, and Computers*, Oct. 29–Nov. 1 1995, Pacific Grove, California, USA.

- "Voiced-Speech Signals Analyzed as a Linear Combination of Single-Sideband Frequency Modulated Components", to appear in the Proceedings of the *Twenty-Ninth Annual Asilomar Conference on Signals, Systems, and Computers*, Oct. 29–Nov. 1 1995, Pacific Grove, California, USA.

5.2 Consultative and Advisory Functions:

None.

5.3 Transitions:

None.

6 New discoveries, inventions or patent disclosures:

None.

7 Honors/Awards:

Elected as a Fellow of the Institute of Electrical and Electronic Engineers.